HeadWize

Projects Library

A DIY Headphone-Amplifier with Natural Crossfeed

by Jan Meier design.zip.

For me, listening to music is a very private affair. Throughout my life, my musical tastes always have been somewhat different from that of the other people I lived with. Consequently, I learned to appreciate good qua "cans." This appreciation has increased since, as a teenager and as a student, I never had the financial means buy loudspeakers that could stand up to the sound quality offered by my Sennheiser and Beyerdynamic headp

Nonetheless, listening was not all heaven on earth. The in-head localization phenomenon did not please me. V recordings presenting a wide soundstage, some instruments are heard in one of the two audio channels only. most annoying, like a bee buzzing in one's ear.

Some 15 years ago I experimented with **electronic** crossfeed by bridging the left and right outputs of the headphone channel of my amplifier with resistors. Although the crossfeed thus produced cured the "buzzing b the sound became extremely dry and I dropped the **idea**.

In the last few years, a number of systems have appeared on the hi-fi market that also produce crossfeed, bu

much more refined way. The analogue headphone amplifiers of HeadRoom and the digital systems of Sony (V 1000), Sennheiser (Lucas), and AKG (Hearo 777) are well-known examples. These systems all more-or-less c problem of the in-head localization experienced with headphones.

These systems work by simulating the mechanisms that a person uses to locate and externalize sources of sou

> First, the sound of a source to the right side of the listener (e.g., the right loudspeaker in a stereo setu only reaches the right ear, but attenuated and delayed, is also heard by the left ear. The level of attenu and the delay time of this crossfeed signal provide important directional information.
>
> Second, the soundwaves are partly absorbed and partly reflected by the listener's head. Especially, the reflections at the ear pinnae interfere with the soundwaves that directly enter the ear canal and amplify attenuate specific frequency components. As these reflections depend on the direction of the soundwav "color" of the sound changes with the direction of the source.
>
> Third, reflections of soundwaves on the walls of the listening room produce reverberation that conveys extra feeling of space.

The information obtained by these mechanisms is further refined by movements of the head. Changes in soun levels, delay times and sound color refine our sense of direction. For a demonstration blindfold a friend and as to locate a ticking clock that you have hidden in the room. He will start turning his head, although he can't see With his head in a fixed position, he will find an exact localization much more difficult.

All these mechanisms are missing when we hear music using headphones. The transducers are directly couple our ears. The sound of the right (left) transducer will not reach the left (right) ear and the reflections on the o have changed and hardly interfere with the original soundwave. Moreover, the sound-sources are attached to head, so head movements no longer add information. Reverberation is not present.

In principle, digital sound processors can simulate the mechanisms described, but the results are, thus far, no satisfactory, because the reflections on the pinnae are very complex and listener-specific.

Fortunately, the mean directional information is provided by the differences between what we hear by our two A "natural" crossfeed from the right (left) audio signal to the left (right) transducer, with an appropriate atten and delay, will reduce most of the adverse symptoms of headphone listening considerably.

A straightforward approach to mimic crossfeed is to take the original stereo signal, attenuate its amplitude an it delayed. Then cross the two channels and add the processed signals to the original stereo signal. In a mathematical formula:

$$V_{left,out}(t) = V_{left,in}(t) + \alpha \, .V_{right,in}(t-t_0) \quad \alpha < 1$$

$$V_{right,out}(t) = V_{right,in}(t) + \alpha \, .V_{left,in}(t-t_0)$$

The HeadRoom systems work like that (with α being a frequency dependent parameter). For a more detailed information just take a look at their most enjoyable homepage (http://www.headphone.com). The people at HeadRoom are very fine engineers, but have not revealed the schematics of their circuitry. So when I decided build my own headphone-amplifier, I had to design an appropriate crossfeed-filter myself.

The crucial part in the crossfeed-filter is the realization of the required time delay of approximately 300 µs. Alt standard solutions for signal delay can be found in many text books on electronics, a fixed frequency-indepen delay with headphones has one major drawback: the so-called Comb-effect.

A conventional crossfeed filter, such as that realized by HeadRoom, mimics the sound of a left or a right soun source most adequately, but the frequency-spectrum of a source in front of the listener is unnecessarily distur For this in-front source, the left and right audio signals are equal: a mono signal. In principle, these signals ne crossfeed. However, with conventional solutions, there still is, and the audio signals at the headphone-transdu become:

$V_{left,out}(t) = V_{right,out}(t) = V_{left,in}(t) + \alpha \, .V_{left,in}(t-t_0) \; \alpha < 1$

Especially in the high frequency range, the delayed crossfeed signal interferes with the original input and atte specific frequencies. The frequency-curve is no longer flat but shows a larger number of dips (the Comb-filter effect). HeadRoom compensates for the overall attenuation with a filter that gently lifts the higher frequencies the dips in the frequency-curve do not disappear.

With a fixed delay, the Comb effect can not be eliminated, so I decided to make the delay of my crossfeed filt frequency dependent. For localization of a sound source, the delays of the frequencies below 2 kHz are the m important, and therefore should have a natural 300 μs delay. For higher frequencies the delay is reduced. Mor by also slightly shifting the original audio signals in the other direction and by giving them a small, frequency dependent attenuation before the crossover signals are added, the mono signals will be left undisturbed! A m signal that simultaneously is found on both channels is left unchanged, and a signal on only one input channel partly transferred to the other channel, with an appropriate time delay at low frequencies.

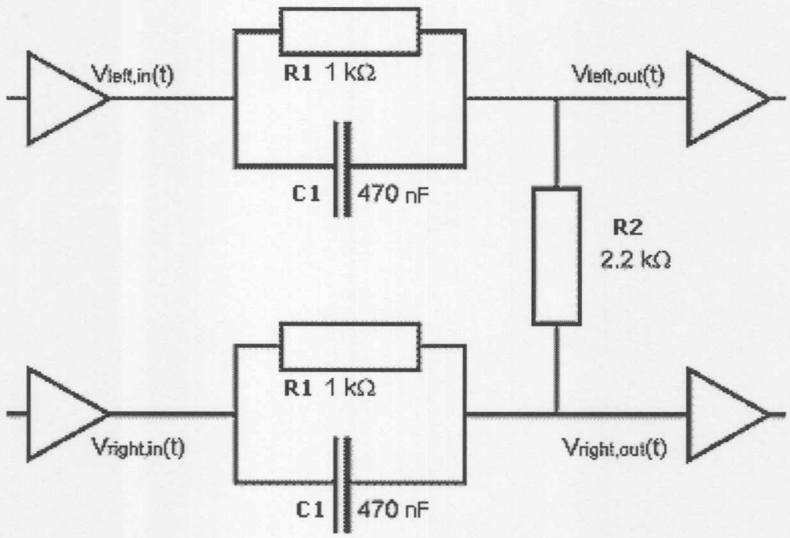


Figure 1

Take a look at the basic crossfeed **circuit** shown in figure 1. The crossfeed is performed by only three resistor two capacitors! It's hard to believe, but the **circuit** really does the job!

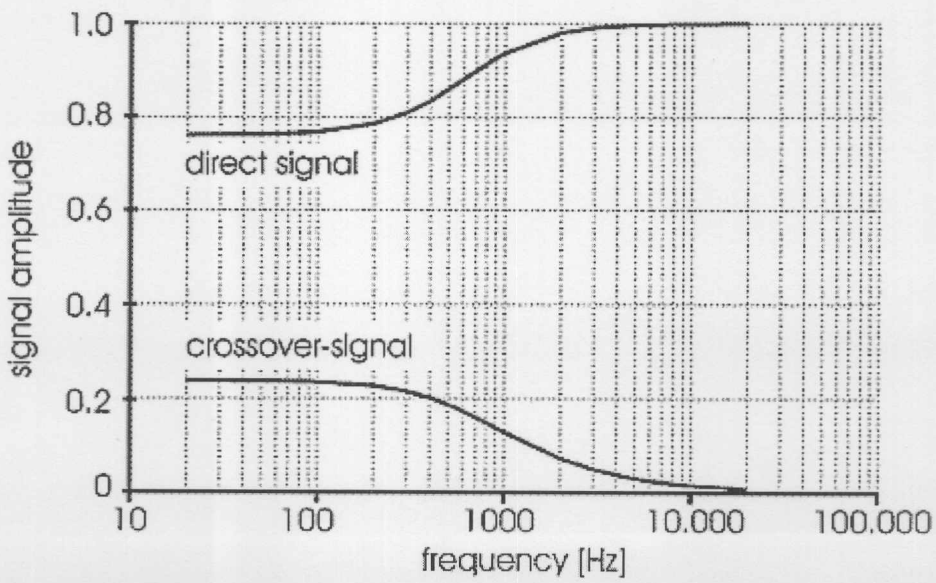


Figure 2

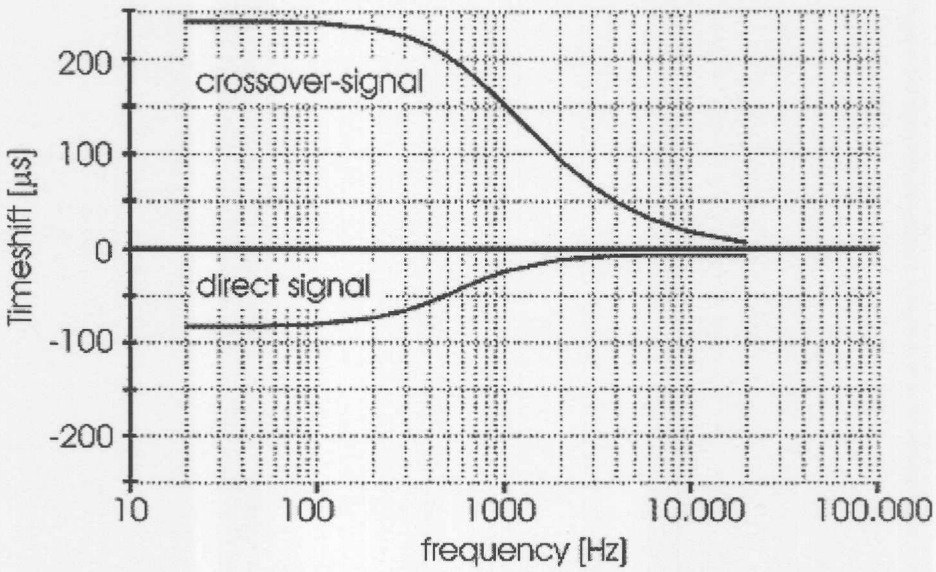


Figure 3

For those interested in the technical details, figures 2 and 3 show the amplitudes and the time delays of both crossfeed-signal as well as the (direct) audio signal. The amplitude of the crossfeed-signal decreases with frec and thus mimics the shadowing effect of the head at higher frequencies. In the lower frequency-range, the tin delay between the crossfeed and the direct signals is 320 µs, and thus mimics the natural delay for a loudspeɛ seen at an angle of approximately 30 degree by the listener. By choosing different parameter values for resist and capacities the crossover signal can be "tuned", but with the values shown it works well for my ears with 9 all my recordings.

SOME MATHEMATICAL STUFF

With a conventional crossfeed filter, the direct signal equals the original input signal, and the crossfeed signal realized by attenuation and delay:

$V_{input}(t) = \cos(2\pi ft)$

$V_{direct} = V_{input}(t) = \cos(2\pi ft)$

$V_{crossover} = a(f).\cos(2\pi f(t-t_{delay}))$

$a(f) << 1$

With a mono signal, both direct signals are equal and both crossfeed signals are equal. The output signals bec

$V_{out}(t) = V_{direct} + V_{crossover} = V_{input}(t) + a(f).\cos(2\pi f(t-t_{delay}))$

The second term interferes with the input/direct signal and results in a number of dips in the frequency spectr those frequencies where: $2\pi ft_{delay} = (2n+1)\pi$. This is the so-called Comb-effect.

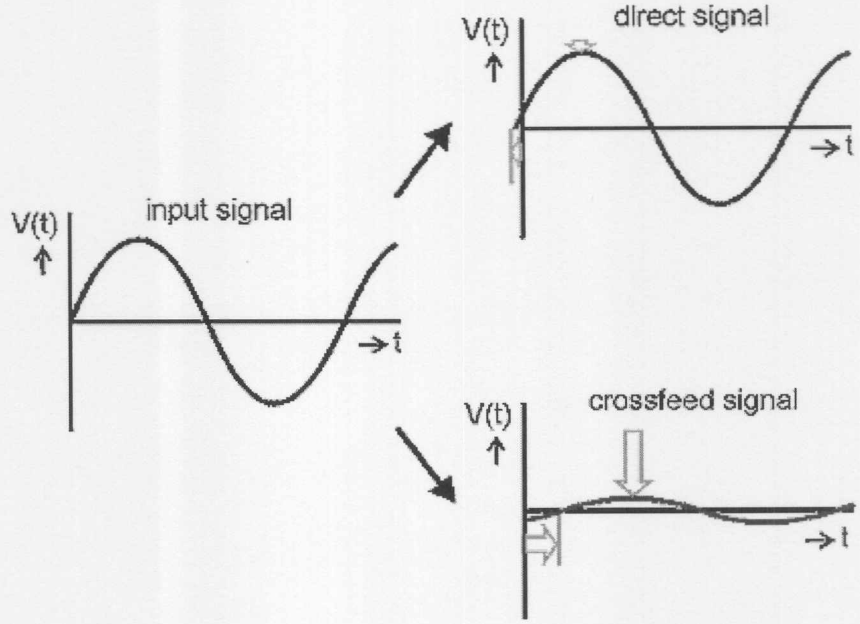


Figure 4

With the "natural crossfeed" filter, the crossfeed signal is also realized by attenuation and delay, but now the signal is also (slightly) attenuated and slightly time-shifted in the other direction (figure 4):

$V_{input}(t) = \cos(2\pi ft)$

$V_{direct} = A(f).\cos(2\pi f(t+t_{direct}))$

$V_{crossover} = a(f).\cos(2\pi f(t-t_{delay}))$

$A(f) = \sin(2\pi ft_{delay})/(\sin(2\pi f(t_{direct}+t_{delay})) \approx < 1$

$a(f) = \sin(2\pi ft_{direct})/(\sin(2\pi f(t_{direct}+t_{delay})) << 1$

$t_{direct} << t_{delay}$

$2\pi f(t_{direct}+t_{delay}) < \pi/2$

(This last condition guarantees that a(f) and A(f) will change monotonically with the frequency.)

The result is that, with a mono signal, the sum of the direct and the crossfeed signals equals the original input and there is no Comb-effect:

$V_{out}(t) = A(f) \cos(2\pi f(t+t_{direct})) + a(f).\cos(2\pi f(t-t_{delay})) = \cos(2\pi ft) = V_{in}(t)$

The condition $2\pi f(t_{direct}+t_{delay}) < \pi/2$ requires that the effective delay ($t_{direct}+t_{delay}$) be shortened for higher frequencies. Natural delay times of 300 µs only can be realized for lower frequencies.

THE NATURAL CROSSFEED NETWORK

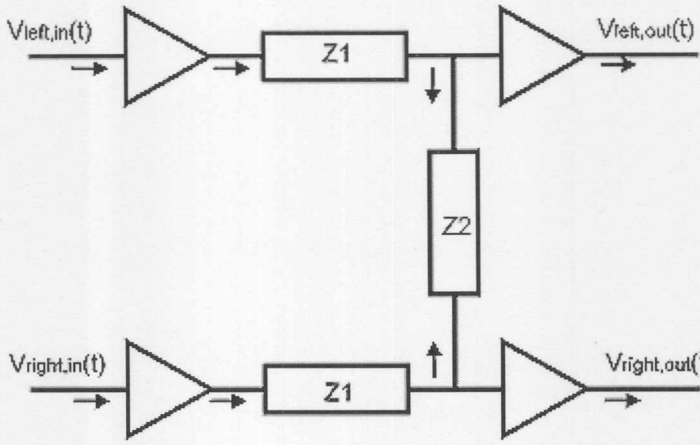


Figure 5

The basic network is shown in the figure 5. It contains a chain of three passive networks, of which the outer t\ have the same impedance value. It easily can be shown that:

$V_{left,out}(t) = (Z_1 + Z_2)/(2*Z_1 + Z_2) * V_{left,in}(t) + Z_1/(2*Z_1 + Z_2) * V_{right,in}(t)$

$V_{right,out}(t) = (Z_1 + Z_2)/(2*Z_1 + Z_2) * V_{right,in}(t) + Z_1/(2*Z_1 + Z_2) * V_{left,in}(t)$

The crossfeed signals are given by the last terms of these two equations.

Using

$Z_1 = R_1 // C_1 = R_1 /(1+i\omega R_1C_1)$

$Z_2 = R_2$

the transfer function of the crossover becomes:

$Z_1/(2*Z_1 + Z_2) = (R_1/(2*R_1 + R_2))/(1+i\omega C_1R_1R_2/(2R_1 + R_2))$

This is the transfer function of a first order low-pass Bessel filter with a filter frequency of:

$f = (2R_1 + R_2)/(2\pi C_1R_1R_2)$

Using:

$R_1 = 1000$ Ohm
$C_1 = 470$ nF
$R_2 = 2200$ Ohm

the filter frequency becomes f = 650 Hz.

To derive the time delay of the crossover signal:

ϕ (phase shift of the transformation) = arctan(ωC1R1R2/(2R1 + R2)) and
τ (time shift) = $\phi/(2\pi f) = \phi/\omega$

For low frequencies:

$\phi = \omega C1R1R2/(2R1 + R2)$
$\tau(\omega \sim 0) = C1R1R2/(2R1 + R2)$

For high frequencies:

$\phi = \pi/2$
$\tau = 1/4f$

To derive the time shift of the direct signal:

Amplitude $A(f) = 1 - Z1/(2Z1+Z2) = (R1+R2+i\omega C1R1R2)/((1+i\omega C1R1R2/(2R1+R2))$

$\phi = \arctan(\omega C2R1R2/(R1+R2)) - \arctan(\omega C2R1R2/(2R1+R2))$

$\tau = \phi/\omega$

$\tau(\omega \sim 0) = C1\ R1\ R2 / (R1 + R2) - C1\ R1\ R2 / (2\ R1 + R2)$

The two filter frequencies are:

$f1 = 1/((2R1+R2)(2\pi C1R1R2) = f$

$f2 = 1/((R1+R2)(2\pi C1R1R2)$

Using the crossover frequency instead, the time delay at low frequencies is $1/(2\pi f) = 250\ \mu s$. Together with a shift of 70 μs of the direct signal, the effective time delay is 320 μs.

MS Excel circuit

Download **circuit** simulation spreadsheet (in MS **Excel** format)

The MS-**Excel** **circuit** simulator (above) lets you experiment with the filter's component values for the standa crossfeed filter (see my article here for a simulator for my enhanced-bass filter). Component values can be ch in the upper left corner of the table. The pictures (the frequency response and time delay characteristics of th crossfeed filter) automatically adapt. The central branch in the spreadsheet consists of a resistor and a capaci series. For the standard version of the crossfeed filter, the capacitance has to be given a very high value so as as a short-**circuit**. Each side-branch has a parallel pair of a resistor and a capacitor in series. Each side branc consists of two resistors and two capacitors.

Circuit model for spreadsheet

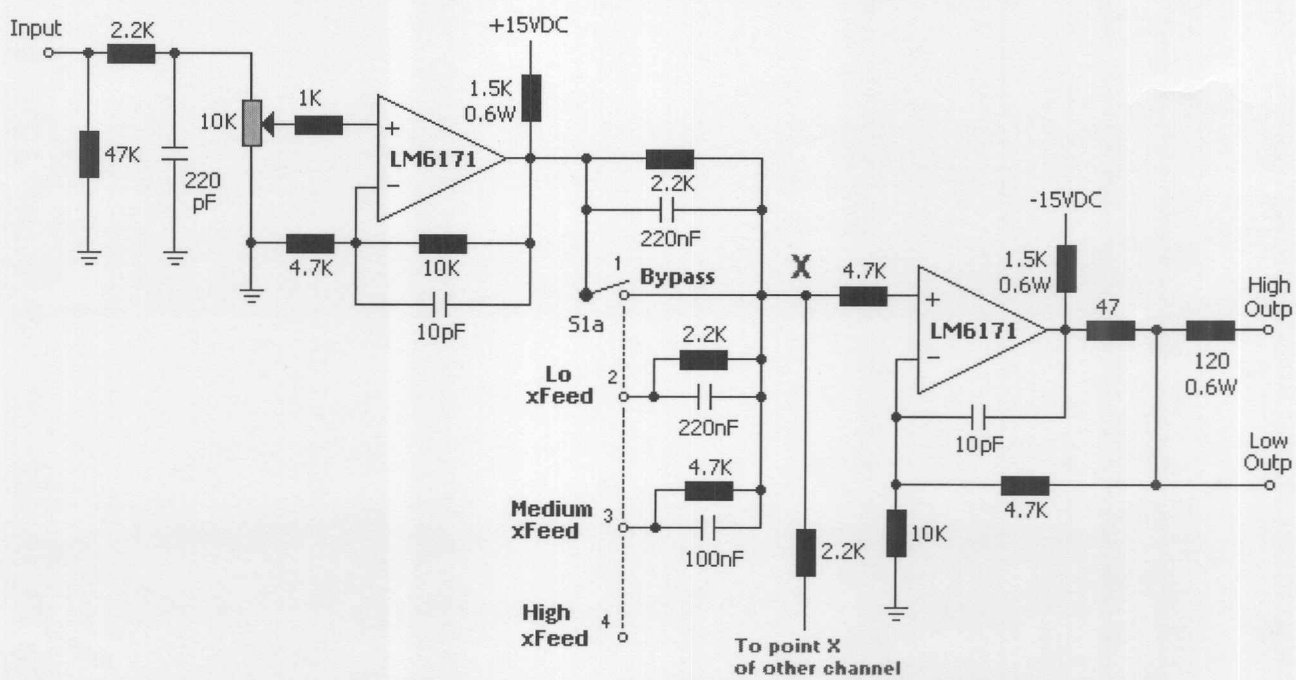
For the standard version of the crossfeed, one resistance has to be set to zero and its corresponding capacitar the regular value (440nF / 320nF / 220nF). The other resistance has the regular crossfeed value (1.1k / 1.5k kOhm) and its corresponding capacitance is set to a very high value. The values in the spreadsheet are for a relatively low crossfeed level.

CONSTRUCTION DETAILS

Building a headphone amplifier is like building a power amp - only the current demand is just a little bit lower a factor 100). Various designs can be found in the internet, and it is relatively easy to integrate the proposed crossfeed filter. I designed my own (see the picture) which is op-amp based. I know, many hi-fi enthusiasts sa

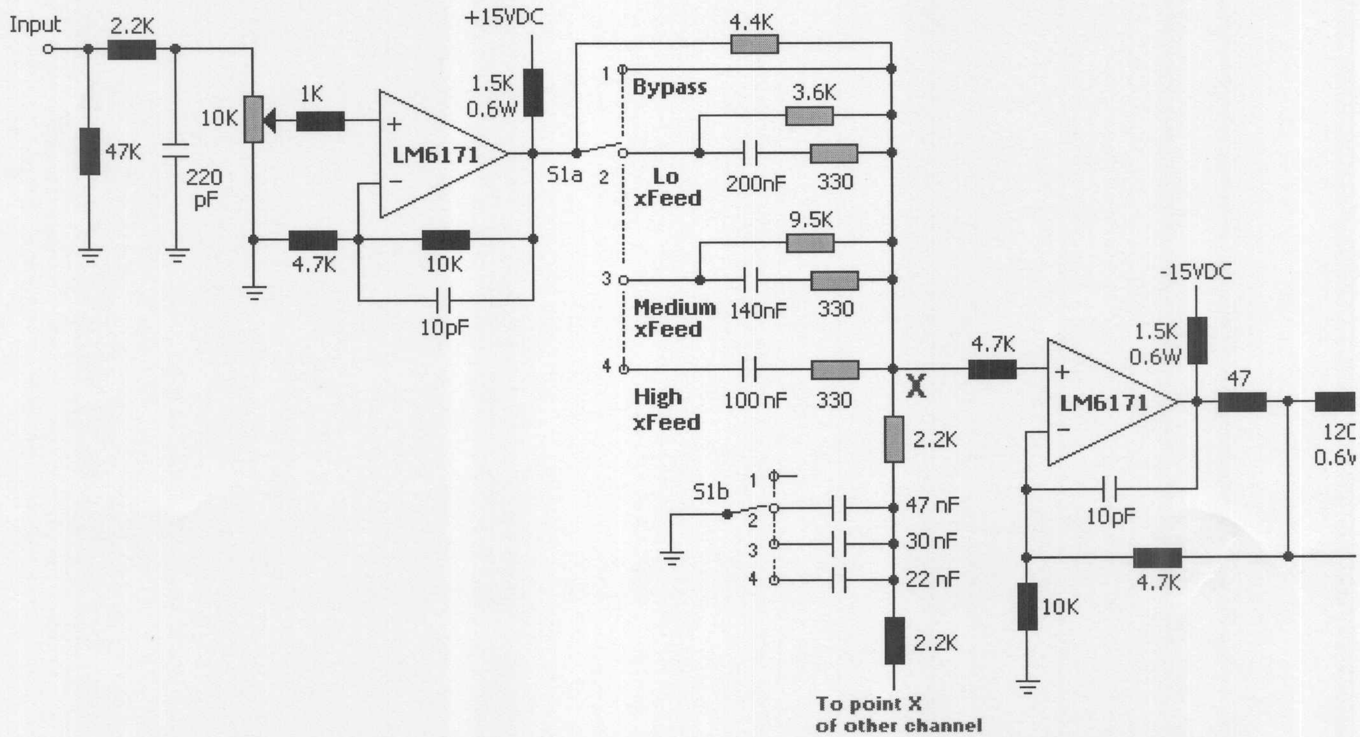
"yuck" to opamps, but note that even in many so-called "High-End" CD-players, opamps are found in the sign
for amplification and filtering.

The op-amps should be chosen with care. They have to be able to deliver relatively high current values and to low impedance loads. I decided to use the National Semiconductor LM6171. This is a wide band (100 MHz) vo feedback opamp that is able to deliver 10V into a 100 ohm load. To prevent difficulties when driving low-impe headphones (32 ohms or less), I placed a 47 ohm resistor at the output of each channel. With my Beyerdynar (DT990/DT931, 600/250 ohm) and my Sennheiser (HD600, 300 ohm) headphones, the opamps perform most adequately. Other alternatives are the LM6172/6181/6182, the OPA604/627 by Burr-Brown (used in the Head systems) or the LTC1206/1207 by Linear Technology (able to drive 30 ohm loads!).



Natural Crossfeed Headphone Amplifier (one channel)

Figure 6a

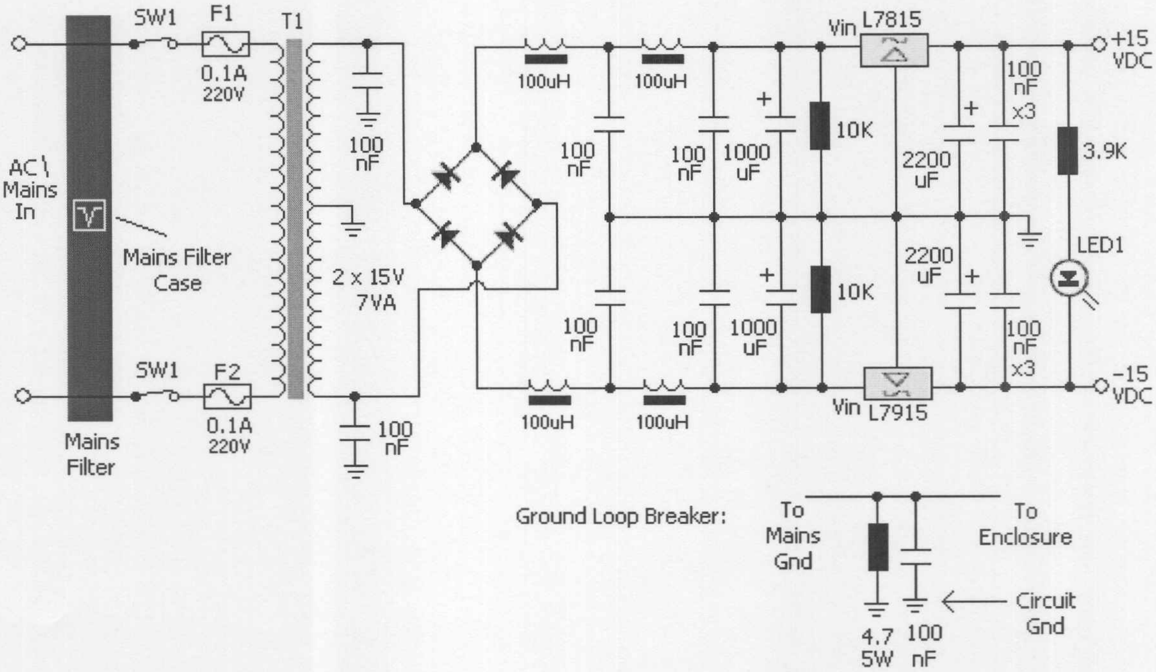


Enhanced-Bass Natural Crossfeed Headphone Amplifier (one channel)
Figure 6b

The schematics shown in figures 6a/6b/6c represent the third generation of my original design. There are two versions of the headphone amplifier: one with the standard crossfeed and one with the enhanced bass crossfe (see An Enhanced-Bass Natural Crossfeed Filter for more information). The standard crossfeed sound is nothir bass-freak. One should not expect a punchy bass, only a relaxation of the sound.

The two crossfeed settings of the original bass-enhanced filter are comparable to the low and the high crossfe levels of the standard filter in this article. The enhanced-bass filter has a frequency response very similar to th modified Linkwitz filter to compensate for any apparent loss of bass due to the crossfeed. While I hardly notic specific loss of bass with the standard filter, I present the enhanced-bass design to give DIYers an option of w filter to build. I have added a medium crossfeed level so that both the standard and bass-enhanced filters nov 4 settings. The 4.4K resistor bridges the switch at all settings to reduce any "blops" during switching.

The output stage of each opamp is connected to one of the voltage rails by a 1.5 kOhm resistor. This forces th output stage into class-A functionality and increases soundquality considerably. Also 10pF capacitors are adde the feedback loop to increase stability at high frequencies. Careful matching of all resistors prevents offset vol and the need of coupling capacitors and the amplifier now is DC-coupled. The power supply has a ground loop breaker, so the audio inputs and outputs MUST have floating grounds - their grounds cannot be directly conne to the enclosure. (See A Precision Preamplifier-Power Amplifier System with Natural Crossfeed Processing for discussion about biasing opamps to function in class A and ground-breakers in power supplies).



Regulated Power Supply for Headphone Amplifier

Figure 6c

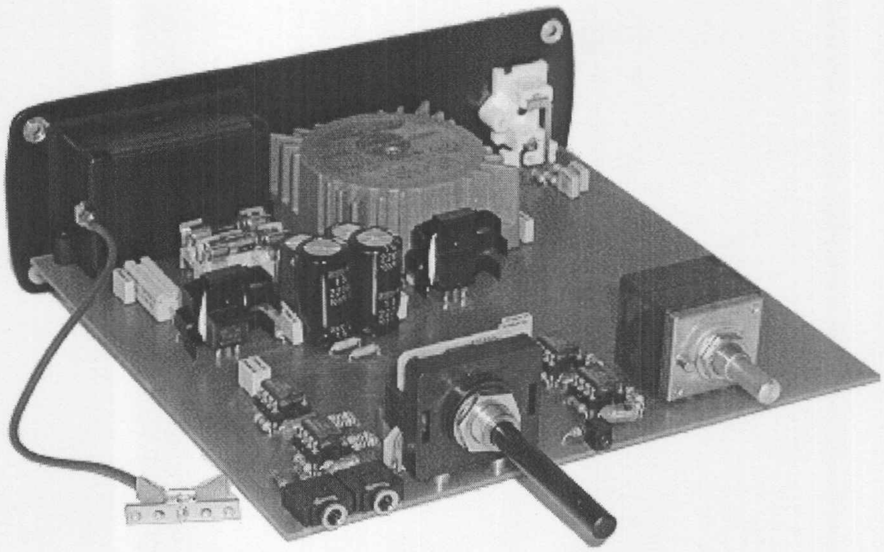
A stable power supply to the opamps is crucial for optimal performance (figure 6c). I used a toroid transforme Charge-reservoirs of 1000uF and two voltage regulators (L7815/7915) provide constant voltages of ±15VDC. Moreover, high frequency noise is extensively filtered by 100mH inductors. 2200uF capacitors and 100nF film capacitors further reduce any ripple and noise after the voltage regulators. I admit that the large capacitors a little bit overdone. When I switch the device off it will still work for about 10 seconds. However, capacitors are relatively cheap, so why not.

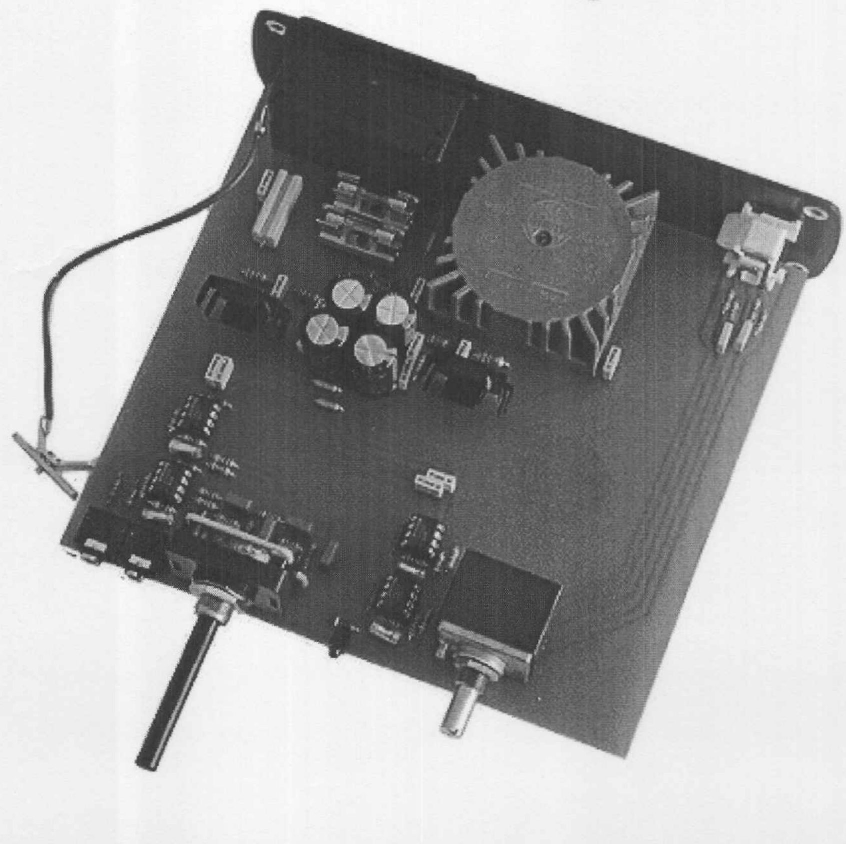
The amplification and power supply circuitry is rather straightforward. I have just a few notes:

Both the standard and enhanced-bass filters have four crossfeed level settings: none (normal stereo m low, medium and high. I personally prefer the low and the medium crossfeed levels for most applicatio

No opamp is perfect. To ensure a zero offset voltage at the output the impedance values in the circuitr been carefully balanced. Do not use other resistor values than as indicated in the schematics, as this m lead to damage of your headphones.

The headamp has two sockets for connection to headphones. Both sockets will provide different sound characteristics. One socket has a very low output impedance and gives the amp tight control over the headphone action. However, many headphones have been sonically optimized to be driven by an outpu impedance of 120 Ohms and may sound better when connected to the other socket. Generally, the low impedance socket provides a clean sound whereas the high impedance socket yields a warmer sound. l one you like most. There is no risk of damage to your headphone by connecting it to either socket. You also use the sockets to connect two headphones simultaneously. However, the volume produced by the impedance socket will be slightly lower than that of the other socket.





As far as the signal paths are concerned, I used WIMA MKS-film capacitors (the red ones), electrolytic capacitors of ELNA, and ¼ Watt metal-film resistors.The 7 Watt toroidal power transformer is by Nuvot Talema, and the 10k logarithmic potentiometer by Alps (the blue one).

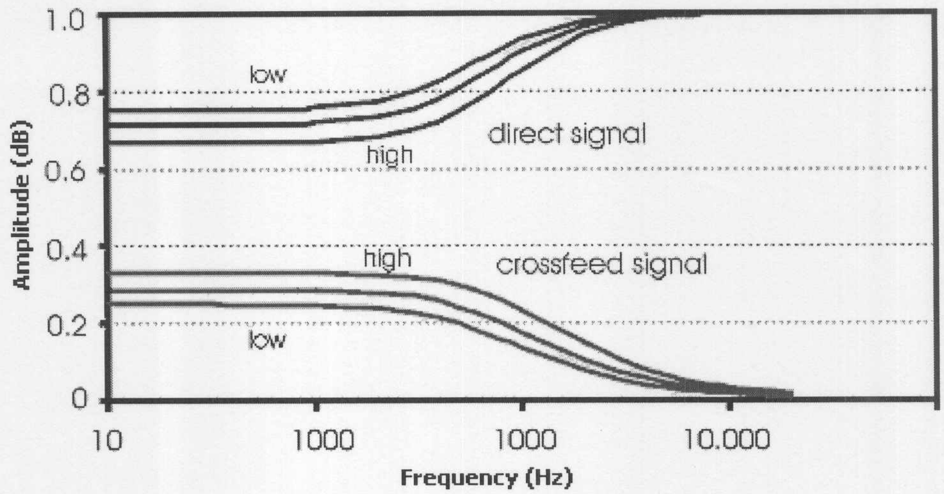


Figure 8

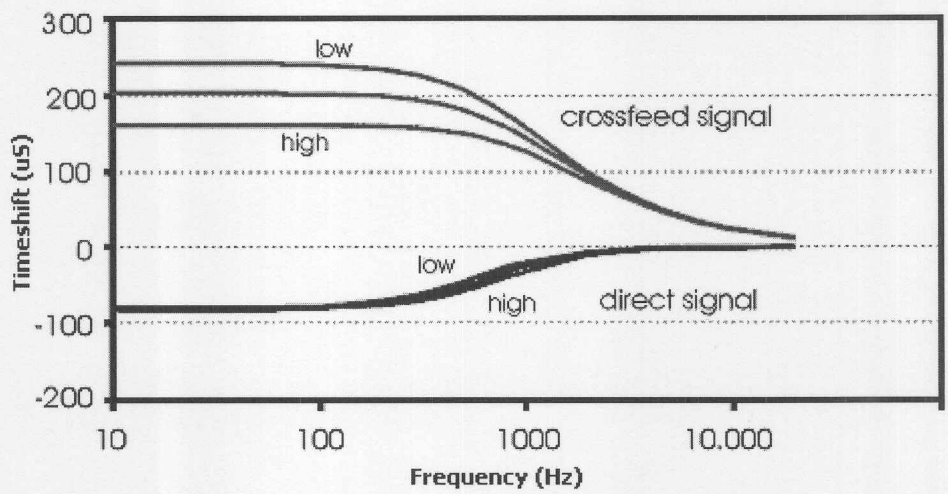


Figure 9

The frequency response and the time-delay of the standard crossfeed network are presented in the figu (figures 8 and 9). Turning the switch from position 1 to position 4 activates and increases the crossfeec personally prefer positions 2 and 3 for most recordings. To see the response graphs of the enhanced-ba filter, see my article here.

Last warning: do yourself a favor and don't use cheap parts. Have a decent (logarithmic) potentiometer or equal), film capacitors, gold plated sockets and a proper metal housing. Even if you don't like and us crossfeed-filter (which I doubt), your headphones are most likely to sound much better than they ever using the normal headphone socket of your CD-player or amplifier. Manufacturers generally do not care the sound quality of headphone sockets and build them cheaply. They sound accordingly.

Now, before you start building, you surely want to know how the crossfeed network performs. I have made 3 recordings to demonstrate the effect of the crossfeed filter. The first recording is made without filtering. The s and third files are made with the filter switch in positions 2 and 4 respectively. The recordings contain a simpl balance test. The effect of the filter on normal music is more subtle and only can be truly appreciated (in my h opinion) while listening for longer periods than is possible with these files.

Balance test w/o filter (1.4Mb - wav)
Balance test w/filter in position 5 (1.4Mb - wav)
Balance test w/filter in position 6 (1.4Mb - wav)

The standard crossfeed amplifier is now commercially available as the Corda HA-1 from Meier Audio. For peop
are interested to build their own headphone amplifier, I'm offering a DIY-kit of the standard crossfeed amplifie
approximately DM 450 ($203 US). I know that it is cheaper to buy all the **electronic** parts by yourself, but pl
note that for the money you have a professional PC board added (with soldering mask, tinned soldering eyes,
the holes drilled) as well as the aluminum case with a 4 mm (!) front- and a 2 mm back-plate with all the hole
milled. Although construction is rather straightforward (a component plan is added), this project is not intende
real novices. If you doubt your own skills please contact the author for the possibilities to obtain a finished de

Well, I never heard any of the HeadRoom systems, but the reviews of these systems very well describe the
impressions that I have with my own amplifier. The sound is not really externalized out of the head, because
reflections and reverberation are missing. However, the soundstage in our head becomes more continuous an
more inner-logic. With some recordings the effect is hardly heard; with other recordings, the effect is rather
extreme. All recordings can be listened to for much longer periods of time, without your head screaming "Tak
headphones off!"

Have fun!

For the latest updates, see the Project Addendum.

The author's website: Meier Audio.
Questions or comments? Visit the HeadWize Discussion Forums.